International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 3 Issue 11, November-2014

Comparison of Bandwidth Utilization in Video Streaming

Abhilasha
Electronics and Communication Engg.,
BIT Mesra

Abstract: **The objective of our work is to estimate the available bandwidth for the video streaming process, so that the optimal utilization of limited bandwidth can be done. Bandwidth estimation is required for the proper utilization of the limited bandwidth. Bandwidth was estimated for Video, Voice, and Data application over the wireless network. The bandwidth estimation in wireless network is more difficult due the interference in the network due the limitation in the radio wave transmission. Our ultimate goal is to provide information to an application, about the maximum bandwidth that is available in the network, so that the application can judge if the network is able to handle the traffic flow without disrupting existing flows in the network and while maintaining acceptable quality of the service.**

1. INTRODUCTION

The term bandwidth has traditionally referred to a static measure of capacity, the maximum amount of data that can be transmitted over a link or path. Available bandwidth is a more difficult quantity to measure since it is a dynamic quantity, the amount of traffic that can be transmitted over a link or path, given current traffic conditions. For an end-to-end path composed of multiple links, the path's bandwidth is limited by the link with the least capacity, referred to as the bottleneck or narrow link.

2. VIDEO STREAMING

Video Streaming is a technique in which sequences of moving images are compressed and are moved over the internet so that the viewers can view the video on receiving it. A complete video streaming system involves all of the basic elements of creating, delivering, and ultimately playing the video content. The main components of complete Video streaming system compromises of Encoding Station, Video Server, Network Infrastructure and Playback Client.

Video streaming system consists of an encoder, distribution server and a client that receives the video data. The distribution server stores the encoded video data and begins to distribute it on the client's demand.

Video Streaming suffers from mainly three basic problems, bandwidth, and jitter and packet loss. These three problems lead to degradation in the quality of the video and time consumed increases as due to delay of packets the time required for buffering increases and if the packet is lost then the quality of the video gets degraded.

2.1 TYPES OF VIDEO STREAMING

Video streaming is basically of two types:

1. Progressive streaming: In the progressive streaming, an ordinary file is first received, and its process is being started before the whole file has been received. In this method of streaming we don't require any special type of protocol. It is processed based on the partial content of the data.
2. True streaming: In the true streaming, the bandwidth of the media signal to the receiver connection is maintained, so that we can see in real time. True streaming is also known as on line streaming as it deals with on line transfer of data.

2.2 BASIC PROBLEMS IN VIDEO STREAMING

Basically there are three problems in the video streaming which affect the performance of the video.

1. Bandwidth: The bandwidth available between two points in the Internet is generally unknown and time-varying. The goal to overcome the bandwidth problem is to estimate the available bandwidth and then match the transmitted video bit rate to the available bandwidth.
2. Delay jitter: The end-to-end delay that a packet experiences may fluctuate from packet to packet. This variation in end-to-end delay is referred to as the delay jitter. Delay jitter is a problem because the receiver must receive/decode/display frames at a constant rate, and any late frames resulting from the delay jitter can produce problems in the reconstructed video, e.g. jerks in the video.
3. Loss rate: The third fundamental problem is losses. A number of different types of losses may occur, depending on the particular network under consideration. Losses can have a very destructive effect on the reconstructed video quality. To combat the effect of losses, a video streaming system is designed with error control.

In order to provide a high-quality video streaming experience for users over a wide range of applications, wireless networks must provide the following essential elements:

1. Sufficient wireless signal:
2. Sufficient wireless bandwidth
3. Quality of service (QoS)
4. Multicast optimization

4. Bandwidth Utilization In Video Streaming

Since raw video consumes a lot of bandwidth, compression is usually employed to achieve transmission efficiency. Video compression can be classified on 2 bases, the video coding technology and scalable video distribution technology. The scalable video distributing technology flexibly overcomes the change in the bandwidth and distributes the video data without any deformation of the image. This technology can automatically adjust the amount of data according to the change in bandwidth.

Encoding and distribution is carried out in real time in the case of live distribution. Load balance is considered by placing the relay server in the appropriate location on the network.

There are several areas that have to be taken care for video streaming. These may include delay, bandwidth allocation, packet loss etc. Among these, delay and delay variation (jitter) are important issues as it may result in degradation of quality of video. Due to this the end user may suffer a poor quality of experience.

There are two ways to transmit the video over the network, **download mode** and **streaming mode.**

The basic idea of video streaming is to split the video into parts, transmit these parts in succession, and enable the receiver to decode and playback the video as these parts are received, without having to wait for the entire video to be delivered.

Video streaming consist of following steps:

- Partition the compressed video into packets.

- Start delivery of these packets.

- Begin decoding and playback at the receiver while the video is still being delivered.

For streaming over the Internet the following protocols are used:

• Media encoding
- MPEG-4 video and audio (AMR for 3GPP), H.263
• Media transport
- RTP for data, usually over UDP/IP
- RTCP for control messages, usually over UDP/IP
• Media session control
-RTSP
• Media description and announcement
- SDP

Advantages:

1. **1.** Audio and video begins playing soon after begin.
2. **2.** Sound quality is good.
3. **3.** Artists and publisher can control distribution and protect copyright.
4. It enables people to conduct real time training session such as webinars with clients or co-workers.
5. Instant play, distributing live events, delivering long-forms of media, multicasting to multiple viewers, and the easy creation of streamed files.

Disadvantages:

1. High cost of server software.
2. Sound quality and stream may be affected by low speed or inconsistent internet connections.
3. Requires a preconfigured server.
4. **4.** Bandwidth availability is the key problem in the delivery of streaming video**.**

4. ARCHITECTURE OF VIDEO STREAMING

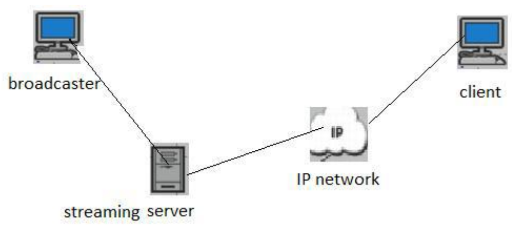
1. Single Sender - Single Receiver Streaming System, is the most common streaming architecture. In this architecture the bandwidth is enough for streaming with constraints of delay and packet recover acknowledgment.

2. Single Sender - Multiple Receivers Streaming System, is a typical broadcast architecture. In this architecture we require bandwidth regulation and adaptation for network conditions.

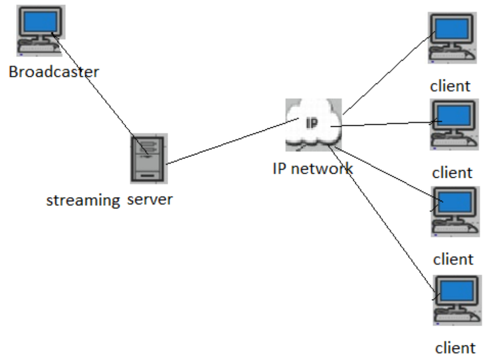
3. Multiple Senders -Single Receivers Streaming System

4. Multiple Senders-Multiple Receiver Streaming System

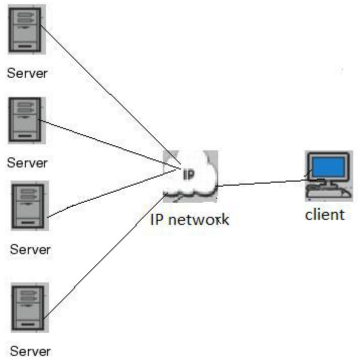
The Multiple sender-Single Receiver and Multiple Sender-Multiple Receiver streaming systems will become popular in the near future because of their distributed system architecture structure. This system should have a robust scheduling structure, because it is necessary to send the media content to client in a certain hierarchical order, and also client should put the received packet in correct order to have the media content.



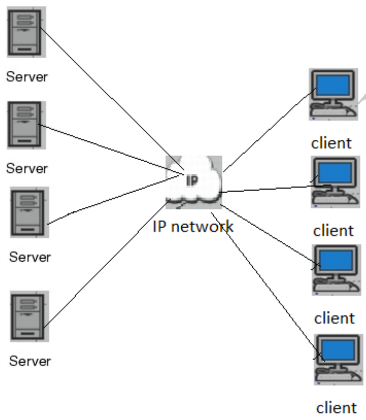
Single Sender - Single Receiver Streaming System



Single Sender and Multiple Receivers Streaming System



Multiple Senders -Single Receivers Streaming System



Multiple Sender- Multiple Receivers Streaming System

5. FACTORS AFFECTING BANDWIDTH UTILIZATION

1. *Network load* is one factor which affects the bandwidth utilization. As the load on the network increases the distortion of data begins. For the data transfer the degradation is less as we increase the data but the bandwidth provided to data application like FTP is comparatively more than the others, so the bandwidth utilization is better in FTP.
2. *Data rate* also affects the bandwidth utilization as with low data rate the utilization is less. As we increase the rate of data the amount of data sent in a particular duration increases, but the amount of bandwidth required is more if we increase the data rate.
3. *Packet Size* plays a important role in determining the utilization of the bandwidth. As we can provide the link with a small packet size and a large packet size also, as in case of small packet size the overload on the network increases due to which the congestion occurs in the network.
4. *Limited Bandwidth* is itself a big problem for bandwidth utilization. As if we estimate the bandwidth and use it and it was under estimated there would be scarcity of bandwidth in that case and the bandwidth is not utilized properly, and if we overestimate also in that case also bandwidth is wasted.

6. TECHNIQUES FOR BANDWIDTH UTILIZATION

1. *Probe Gap Model*= PGM compares the time gap of successive probe packets between the sender and the receiver to calculate the available bandwidth. This dispersion is used to estimate the amount of cross traffic in the link during time T which is required to be subtracted from the capacity to estimate the available bandwidth in the path. The main component of PGM is a mathematical relation between the input and output rates of a probing packet pair, under the fluid traffic control model. The calculation PGM tool is usually based on a set of congested probe packets.
2. *Probe Rate Model*= Probe Rate Model is based on the concept of induced congestion. In this technique available bandwidth is determined by the variation in the probing packet rate from sender to the receiver. PRM methods exploited self-induced congestion to detect available bandwidth. Specifically, when the traffic probing rate is larger than the available bandwidth, the queue at the bottleneck link begins to grow such that probe packet is forced to be delayed. The probe packet begins to increase its delay once the probe traffic rate is over the turning point. At this moment, the available bandwidth is defined to be the transmission rate of probe traffic at the turning point. However, PRM needs to pour more packet pairs to obtain reliable estimation such that it incurs intrusiveness and needs long convergence time.

7. CONCLUSION AND FUTURE WORK

As we increase the load, or we increase the resolution of video or the size of packets then the required amount of bandwidth also increases. If the required bandwidth is insufficient then there is loss of data and the quality of the video also degrades.

In future we can implement an IP cloud in the existing scenario regardless the amount which is used to install the IP cloud. With the help of IP cloud we can adjoin bigger Scenarios which can bear the load of the transmission and the reception. We can also implement the switch and a router for a better transmission of the data and so that bandwidth can be better utilized.

8. REFERENCES:

1. S. Bellovin. A Best-Case Network Performance Model. Tech. Rep., ATT Research, Center for Research and Scholarships, Columbia University, Feb. 1992. http:// academiccommons.columbia.edu.
2. Carter, R.L. and Crovella, M.E., "Dynamic Server Selection using Bandwidth Probing in Wide-Area Networks", Technical Report, computer Science Department, Boston University, March 1996. http://hdl.handle.net/2144/1584.
3. V. Jacobson. Pathchar: A tool to Infer Characteristics of Internet Paths. ftp://ftp.ee.lbl.gov/pathchar/, April 1997. Viewing Date: 10th November 2013 Viewing Time:15:24.
4. Kevin Lai and Mary Baker, “Measuring Bandwidth,” Proceedings of IEEE Conference on Computer Communications (INFOCOM), New York, NY, USA, Apr. 1999,pp. 235–245.
5. F.P. Nicolai, “Capacity measurement in small-scale heterogeneous best-effort IP networks” thesis, Delft University of technology, 2000.
6. Jacob Strauss, Dina Katabi, and Frans Kaashoek. “A Measurement Study of Available Bandwidth Estimation Tools”, Proceedings of the ACM SIGCOMM Internet Measurement Conference ’03, Miami, Florida, October 2003.
7. Rohit Kapoor, Ling-Jyh Chen, Li Lao, Mario Gerla, and M. Y. Sanadidi, “Capprobe: a simple and accurate capacity estimation technique,” *SIGCOMM* Comput. Commun. *Rev.*, vol. 34, no. 4, pp. 67–78, 2004.
8. Karthik Lakshminarayanan, Venkata N. Padmanabhan, and Jitendra Padhye,
“Bandwidth estimation in broadband access networks.,” Proceedings of IMC *2004*, Taormina, Sicily, Italy, Oct. 2004.
9. Yu-Chen Huang, Chun-Shien Lu1, Hsiao-Kuang Wu ,“Reliable Available Bandwidth Estimation Based on Distinguishing Queuing Regions and Resolving False Estimations” Institute of Information Science, Academia Sinica, Taipei, Taiwan, ROC Dept. Computer Sci. & Information Eng., National Central Uni., 78 Chung-Li, Taiwan, ROC, IEEE, 2004.http://www.iis.sinica.edu.tw/papers/lcs/1527-F.pdf.
10. Samarth H. Shah, Kai Chen, Klara Nahrstedt ,“Available Bandwidth Estimation in IEEE 802.11-based Wireless Networks” Department of Computer Science University of Illiniosis at Urban-Champaign. http://www.caida.org/workshops/isma/0312/abstracts/shs